IN THE CLAIMS

1. (currently amended) An end-to-end estimation of the bandwidth available in a client-server connection established over a packet switching network, comprising:

a routine to compute samples of available bandwidth by taking into account the flow of packets received by the client and their arrival times, if the routine is implemented at the receiver side, or by taking into account acknowledgment or report packets received by the sender and their arrival times, if the routine is implemented at the sender side;

a routine to compute bandwidth samples as the ratio of the amount of received data packets over the time interval during which data packets are received if the routine is implemented at the receiver side, or as the ratio of the amount of data packets acked over the time interval during which data are acked if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth.

2. (currently amended) The end-to-end bandwidth estimation according to claim 1, wherein a sample of available bandwidth b_j at time t_j is computed as:

$$b_j = \frac{d_j}{t_j - t_{j-1}}$$

where d_j is the amount of data that have been received at the receiver or acknowledged at the sender in the interval $t_j - t_{j-1}$, t_{j-1} is the time when the previous ACK or the ACK one congestion window of packets before was received by the sender or the time when the previous packet or the packet one congestion window of packets before was received by the

receiver, and t_j is the time when the current ACK is received by the sender or when the current packet is received by the receiver.

3. (original) The end-to-end bandwidth estimation according to claim 1, wherein the routine implements a discrete time low-pass filter with time-varying coefficients.

4. (currently amended) The end-to-end bandwidth estimation according to claim ~~1~~ 2, wherein the available bandwidth samples are computed according to claim 2 and are averaged using ~~routine implements~~ the discrete-time low-pass filter with time-varying coefficients:

$$\hat{b}_j = \frac{2\tau_f - \Delta_j}{2\tau_f + \Delta_j}\hat{b}_{j-1} + \Delta_j \frac{b_j + b_{j-1}}{2\tau_f + \Delta_j}$$

where $\hat{b}_j$ is the filtered measurement of the available bandwidth at time $t = t_j$, $\hat{b}_{j-1}$ is the filtered measurement of the available bandwidth at time t_{j-1}, $\Delta_j = t_j - t_{j-1}$, $1/\tau_f$ is the cut-off frequency of the filter, b_j is the sample of the available bandwidth at time t_j, and b_{j-1} is the sample of the available bandwidth at time t_{j-1}. If a time τ_f/m ($m \geq 2$) has elapsed since the last received ACK or packet without receiving any new ACK or packet, then the filter assumes the reception of a *virtual* sample $b_j=0$.

5. (original) Method for adapting the amount of data for unit of time, i.e. the rate, sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation according to claim 1.

6. (previously presented) Method for adapting the amount of data for unit of time sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth,

wherein the low-pass filter is a low-pass filter according to claim 3.

7. (currently amended) Method for adaptively setting congestion window (cwnd) and slow start threshold (ssthresh) in the TCP/IP protocol[[,]] comprising an end-to-end bandwidth estimation according to claim 1 to set the windows as follows:

after a timeour: ssthresh=BWE*RTTmin, cwnd=2;

after 3 dupack: ssthresh=BWE*RTTmin, cwnd=ssthresh; and

where RTT min is the minimum round trip time and BWE is the available bandwidth computed according to claim 1 at the time of timeout or when 3 dupacks are received.

8. (previously presented) Method for adaptively setting congestion window and slow start threshold in the TCP/IP protocol comprising an end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth,

wherein the low-pass filter is a low-pass filter according to claim 3.

9. (original) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source using the TCP protocol, or the UDP protocol or the RTP protocol, comprising an end-to-end bandwidth estimation according to claim 1.

10. (previously presented) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source using the TCP protocol, or the UDP protocol or the RTP protocol comprising and end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth over a packet network, comprising an end-to-end bandwidth,

wherein the low pass filter is a low pass filter according to claim 3.

11. (previously presented) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source according to claim 9, comprising:

increasing step by step the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source until congestion is experienced by means of control packets;

setting the quality of coding or select the numbers of layers to be transmitted after that a congestion episode is signaled by means of control packets in according with the bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side;

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth over a packet network, comprising an end-to-end bandwidth and;

increasing again step by step the quality of coding or the numbers of layers to be transmitted in a layered coding to probe for extra available bandwidth.

12. (currently amended) Method for setting the Advertised Window of TCP equal to the minimum of the Advertised Window and the bandwidth estimate times the minimum round trip time, wherein the ~~samples are~~ bandwidth is computed according to claim ~~1, and the bandwidth estimate is computed by:~~

~~a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and~~

~~a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth over a packet network, comprising an end-to-end bandwidth~~ 1.

13. (previously presented) Method for adapting the amount of data for unit of time sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth,

wherein the low pass filter is a low pass filter according to claim 4.

14. (previously presented) Method for adaptively setting congestion window and slow start threshold in the TCP/IP protocol comprising an end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth,

wherein the low-pass filter is a low-pass filter according to claim 4.

15. (previously presented) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source using the TCP protocol, or the UDP protocol or the RTP protocol comprising and end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth over a packet network, comprising an end-to-end bandwidth,

wherein the low-pass filter is a low-pass filter according to claim 4.

16. (previously presented) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source according to claim 10, comprising:

increasing step by step the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source until congestion is experienced by means of control packets;

setting the quality of coding or select the numbers of layers to be transmitted after that a congestion episode is signaled by means of control packets in according with the bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side;

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth over a packet network, comprising an end-to-end bandwidth; and

increasing again step by step the quality of coding or the numbers of layers to be transmitted in a layered coding to probe for extra available bandwidth.

17. (previously presented) Method for setting the Advertised Window of TCP equal to the minimum of the Advertised Window and the bandwidth estimate times the minimum round trip time, wherein the samples are computed according to claim 2, and the bandwidth estimate is computed by:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth over a packet network, comprising an end-to-end bandwidth.

18. (new) Method for adapting the amount of data for unit of time, i.e. the rate, sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation according to claim 2.

19. (new) Method for adapting the amount of data for unit of time, i.e. the rate, sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation according to claim 3.

20. (new) Method for adapting the amount of data for unit of time, i.e. the rate, sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation according to claim 4.